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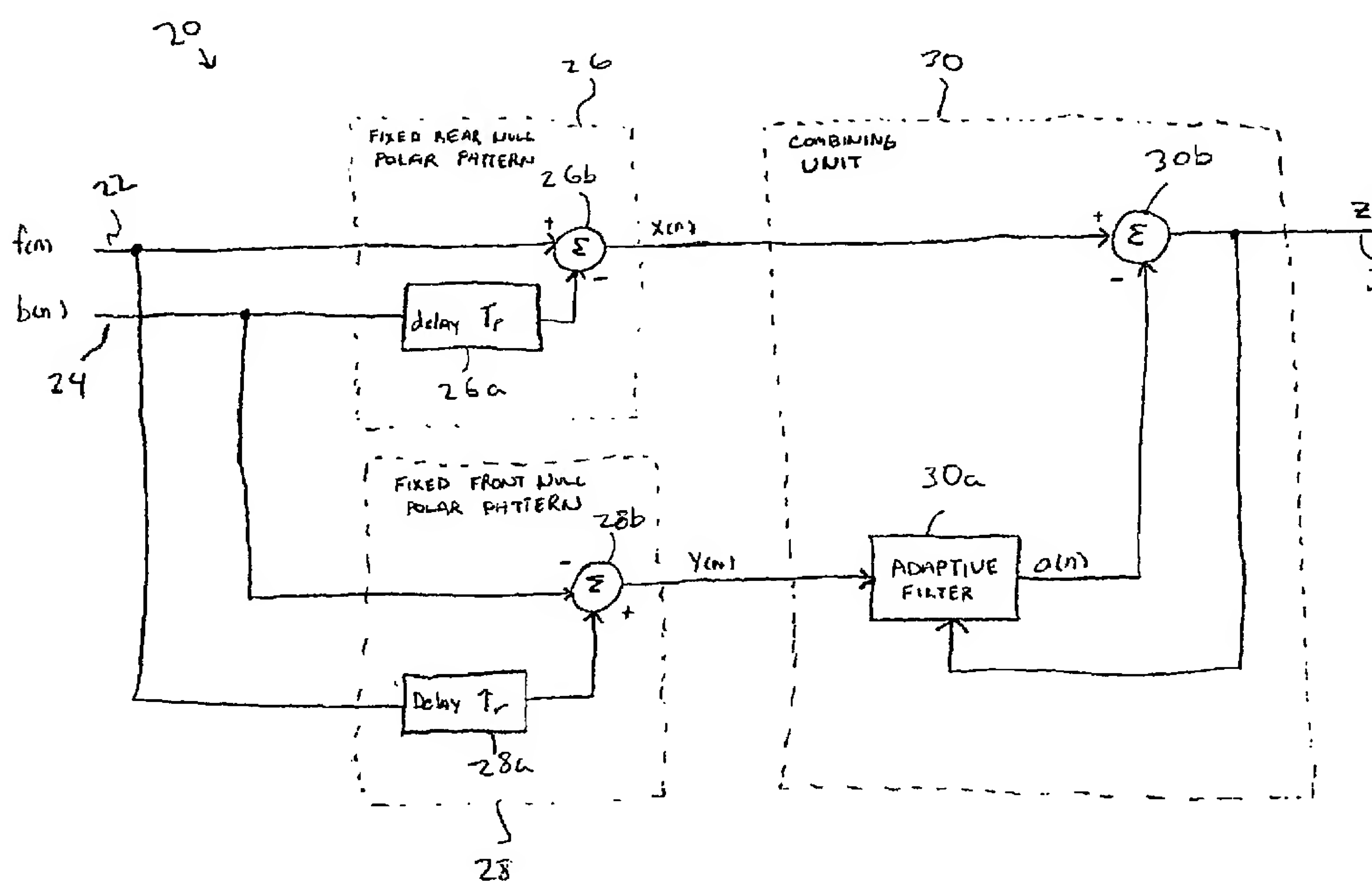
(54) Title: FIXED POLAR-PATTERN-BASED ADAPTIVE DIRECTIONALITY SYSTEMS



(57) Abstract: A dual fixed polar pattern based adaptive directional system uses a fixed rear-null polar pattern unit to produce an enhanced-speech value, and a fixed front-null pattern unit to produce an enhanced noise value. The enhanced noise value is adaptively filtered and the filtered values combined with the enhanced signal values to produce the output for the microphone system. Since only the enhanced noise signal is provided to the adaptive filter and the adaptive filter is updated adaptively by minimizing the output power of the system, the sytem will provide the noise-minimized output.

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FIXED POLAR-PATTERN-BASED ADAPTIVE DIRECTIONALITY SYSTEMS

Background of the Invention

The present invention relates to adaptive-directionality microphone systems.

In many situations it is useful to combine the output of multiple antenna elements to produce an antenna system with directionality. For example, when a microphone system is associated with a user, the desired speech signal typically comes from the front, while the noise tends to be ambient or from a direction other than the front.

Fig. 1 illustrates a prior-art microphone system using a fixed delay. The output of the front microphone is sent to combiner 12 while the output of the back microphone is sent to a delay 14 and then to the combiner 12. It can be shown that if d is the distance between the two microphones, and the delay is set equal to $\frac{d}{c}$, where c is the speed of sound, a cardioid polar pattern output is produced by the system of Fig. 1. This cardioid polar pattern has a null at 180°. Such fixed null systems do produce the improved signal to noise ratio. The target speech signal from the front is enhanced over the ambient noise. The delay 14 can be set to other values in which case nulls are created at other angles. One disadvantage of fixed delay systems is that it doesn't allow for adaptive directionality. In some cases it is desired that a null can track a noise signal to produce an improved signal-to-noise ratio.

Fig. 2 is a diagram of one adaptive directionality system. In this figure, the output of the combiner 18 is sent to control the adaptive delay unit 19 to modify the delay so that the nulls can track the noise signal. One disadvantage of the system of Fig. 2 is that when the desired speech signal is greater than the noise signal, the microphone system of Fig. 2 can move the nulls toward the front, where the speech signal is located.

Many techniques for both fixed modes and adaptive modes have been suggested. However, most of the techniques can not be practically implemented in hearing aids because of the limits of hardware size, computational speed, mismatch of microphones, power supply, and other factors. The most common technique used in hearing aids is a directional microphone or dual omnidirectional system with a fix polar pattern, as shown in Fig. 1.

There is an increasing demand for hearing aids with adaptive directionality, but the practical factors mentioned above prevent the current adaptive directionality algorithms from being practically used. First, the performance of any adaptive directionality scheme greatly depends upon the distance and number of microphones. However, common hearing aids such as behind-the-ear hearing aids can only have two microphones, and the distance between the microphones is limited to about ten millimeters. This means that the corresponding system can effectively cancel only one noise source. In the case of multiple noise sources, or diffuse noise, most of the available adaptive directionality schemes with only two closely spaced microphones will deliver a very poor performance. Secondly, as discussed above, there is a serious target signal cancellation problem, which means that the system cancels not only the noise but the target signal if the assumptions of the directionality system, such as the relative strengths of the target signal and noise is not exactly matched.

It is desired to have an improved adaptive directionality system, especially a microphone system with adaptive directionality for use in hearing-aid applications.

<u>Summary of the Present Invention</u>

The present invention is a system that uses two fixed polar-pattern units, and combines the outputs of these two units to produce an adaptive directionality output.

In a preferred embodiment, the first fixed null unit is a fixed rear-null polar pattern unit which produces an enhanced speech signal output. The second fixed null unit is a fixed front-null polar pattern unit. This fixed front-null polar pattern unit produces an enhanced noise signal. The enhanced noise signal can be combined with the enhanced speech signal to produce an adapted directionality output in a combining unit. Preferably the output of the enhanced noise signal is sent to an adaptive filter and then combined in a summer with the enhanced speech signal output. The functional units of the present invention can be implemented in a digital signal processor.

In a preferred embodiment, the system output is used to adjust the adaptive filter to minimize the expectation value of the output power. Since the fixed front-null polar pattern output sent to the adaptive filter does not specifically depend upon the front speech signal, the system of the present invention will not tend to move a null towards the front.

Brief Description of the Drawings

Fig. 1 is a diagram of a prior-art fixed delay directionality microphone system.

Fig. 2 is a diagram of a prior-art adaptive delay directional microphone system.

Fig. 3 is a diagram of one embodiment of the adaptive directionality microphone system of the present invention.

Fig. 4A is a diagram of a polar pattern for a fixed rear-null polar pattern unit used in one embodiment of the microphone system of the present invention.

Fig. 4B is a diagram of a polar pattern for a fixed front-null polar pattern unit used in one embodiment of the microphone system of the present invention.

Fig. 5 is a diagram that illustrates one implementation of the microphone system of the present invention.

Fig. 6 is a flow chart that illustrates the operation of one embodiment of the microphone system of the present invention.

Detailed Description of the Preferred Embodiment

Fig. 3 is a diagram illustrating one embodiment of the microphone system of the present invention. The microphone system includes data from a front microphone on line 22 and from a rear microphone on line 24. The signals for the front and rear microphones are supplied to a fixed rear-null polar pattern unit 26 and a fixed front-null polar pattern unit 28. The outputs of the fixed rear-null polar pattern unit 26 and the fixed front-null polar pattern unit 28 are sent to a combining unit 30 that produces the output on line 32. In a preferred embodiment, the fixed rear-null polar pattern unit 26 includes a delay unit 26a and a combining unit 26b. The fixed front-null polar pattern unit 28 includes a delay unit 28a and a combining unit 28b.

Fig. 4A illustrates the polar pattern of the output of a fixed rear-null polar pattern unit 26. This shows a rear-null cardioid pattern. Note that the fixed rear-null polar pattern unit produces an enhanced-speech signal output because the null at 180° reduces the noise signal. Fig. 4B illustrates the polar pattern of the output of a fixed front-null polar pattern unit. This pattern has a null at zero degrees so the output of the fixed front-null polar pattern unit 28 is an enhanced noise signal with the speech signal from the front being strongly filtered.

Note that although Figs. 4A and 4B illustrate a cardioid pattern, other patterns such as hypercartioids or supercardioids can also be used.

Looking again at Fig. 3, both the enhanced speech signal $x(n)$ and the enhanced noise signal $y(n)$ are provided to the combining unit 30. In one embodiment, an adaptive filter 30a receives the enhanced noise signal. The

filtered output $a(n)$ is then produced and subtracted in the summer 30b from the enhanced speech signal $x(n)$ to produce the output $z(n)$.

The output signal is used by the adaptive filter 30a to modify the coefficients of the adaptive filter 30a. In one embodiment, the expectation value of $z^2(n)$ is minimized. Since the enhanced noise signal is provided to the adaptive filter, the adaptive directionality microphone system of Fig. 3 will not move its nulls toward the target speech signal. That is because the enhanced noise signal provided to the adaptive filter 30a does not include significant components produced by the speech signal. The filter instead attempts to adjust itself so that the noise portion is filtered to minimize the expectation value of the output power. In one embodiment, the adaptive filter 30a is a finite impulse response (FIR) filter.

One embodiment of the present invention is described with respect to a mathematical example. The received signals at the front microphone and rear microphone are $f(n)$ and $b(n)$; τ_p and τ_r are selected to be equal to t and t is equal to $\frac{d}{c}$; d is the distance between two microphones, c is the speed of sound. As described below, other patterns can be used. The signals $x(n)$ and $y(n)$ are called the primary signal and the reference signal, respectively, as used in Griffiths-Jim type algorithms; $z(n)$ is the output of the system; $W(n) = [W_1(n), W_2(n), \ldots, W_N(n)]^T$ is the weight vector of the Nth-order adaptive FIR filter; $a(n)$ is the output of the adaptive filter. Their relationships are as follows:

$$f(n) = s(n) + i(n) \tag{1}$$

$$b(n) = s(n - \tau) + i(n - \tau_1) \tag{2}$$

$$x(n) = f(n) - b(n - \tau_p) \tag{3}$$

$$y(n) = f(n - \tau_r) - b(n) \tag{4}$$

$$a(n) = \sum_{m=1}^{N} W_m(n) y(n - m + 1) = W^T(n) Y(n) \tag{5}$$

$$z(n) = x(n) - a(n) \tag{6}$$

where $s(n)$ and $i(n)$ are the desired signal part and the noise part in the front microphone, respectively; τ_I is the delay of the noise transmission from the front microphone to the rear microphone and is equal to $\frac{d}{c}cos(\theta)$, θ is the angle of the noise along the line between two microphones; $Y(n) = [y(n), y(n-1), \ldots\ldots, y(n-N+1)]^T$ is the input of the adaptive filter. In the above, we also assume that the desired signal comes from straight ahead, that is, its angle is zero. It is easy to see that in the ideal case $y(n)$ contains only the noise part. The adaptive filter can provide by use of some learning algorithms an output $a(n)$ approximately equal to the noise part in the primary signals $x(n)$. As a result, the system output $z(n)$ is an approximate noise-free signal. From a polar pattern point of view, this means that the null of the polar pattern will adaptively be towards the direction of the noise by

adaptively updating the weights of the filter. For example, in the case of $N=1$, if the updating weight $W_1(n) = -0.35$, then the null at the time n will be at about 120° and 240°; if $W_1(n) = -0.5$, then the null at the time n will be at about 110° and 250°, etc. In a preffered embodiment, the coefficient is limited to the range 0 to -1 so that the null will range from 90° to 180° and 180° to 270°.

Concerning the learning algorithms, we can directly use some available adaptive algorithms such as LS, RLS, TLS and LMS algorithms. The LMS algorithm version to update the coefficients of adaptive filter is

$$W(n+1) = W(n) + \lambda Y(n) z(n) \tag{7}$$

where λ is a step parameter which is a positive constant less than $\frac{2}{P}$ and P is the power of the input of the adaptive filter. For better performance and faster convergence speed, λ can be also time varying as the normalized LMS algorithm uses, that is,

$$W(n+1) = W(n) + \frac{\mu}{\|Y(n)\|^2} Y(n) z(n) \tag{8}$$

where μ is a positive constant less than 2.

Based on the frame-by-frame processing configuration , a further modified algorithm can be obtained as follows:

$$W_k(n+1) = W_k(n) + \frac{\mu}{\|Y(n)\|^2} Y(n) z_k(n) \tag{9}$$

where k represents the $k'th$ repeating in the same frame.

In one embodiment, the system of Fig. 3 is implemented in a digital signal processor. Fig. 5 illustrates one embodiment of a digital implementation of the system of Fig. 3. The adaptive directionality system 50 includes a front microphone 52 and a rear microphone 54. The microphones are sent to analog-to-

digital (A/D) converters 56 and 58. The digital signals are then provided to the processor 60. The processor 60 operates on the digital microphone samples by running the "dual fixed polar pattern-based adaptive directionality program" 62 which implements the system of Fig. 3. The delaying, summing and filtering as shown in Fig. 3 are all implemented digitally.

Fig. 6 is a flow chart that illustrates one implementation of a dual fixed-pattern-based adaptive directionality program implemented using a digital signal processor (DSP). In step 70, samples of the front and back microphones are used to calculate the output of a rear-null fixed polar pattern block. This produces an enhanced speech signal output. This step can be implemented by delaying samples from the rear microphone by the time τ_p and subtracting the delayed values from the values received from the front microphone. In step 72, samples of the front and back microphones are used to calculate the output of the front-null fixed polar pattern block. This is the enhanced noise signal output. In one embodiment, the delay τ_r is given to the samples from the front microphone, and then the samples from the rear microphone are subtracted from this delayed value. In step 74, the output samples of the front-null fixed polar pattern block (the enhanced noise signal) are adaptively filtered to produce an adaptive filter output $a(n)$. In a preferred embodiment, the adaptive filter acts by modifying the weighting coefficients using current and previous values of the enhanced noise signal $y(n)$ to implement a finite impulse response filter. In step 76, $x(n)$ value is combined with the output of the adaptive filter to produce the system output. In this embodiment the output $z(n)$ is equal to $x(n) - a(n)$. In step 78, the system output value is used to modify the filter coefficients for the adaptive filter. This can be done using a variety of different adaptive algorithms.

A number of modifications can be made to the above embodiments. For example, the delays τ_p and τ_r are used to provide the fixed polar pattern. This fixed polar pattern can be a variety of other polar patterns such as the

supercardioid or hypercardioid. The delay portions can be implemented during the analog-to-digital conversion stage or by the use of a corresponding digital filter such as an all-pass filter or an FIR filter which has a fractional-sample delay property. The adaptive algorithms for the adaptive filter can be any least-means-squared (LMS) based, LS-based, TLS-based, RLS-based or related algorithms. In one embodiment, a single coefficient filter is used. The weights can also be obtained by solving the estimated Wienner-Hopf equation. Repeated adaptive algorithms like Equation 9 or an adaptive lattice filter can be used in this scheme as well. The length of the adaptive filter can be adjustable. Trade-offs between performance and cost (computational complexity, etc.) will help determine the algorithm in practical applications.

A matching filter can also be added in either of the dual microphones before or immediately after performing the delay processing so as to compensate for the magnitude mismatch of the two microphones. The matching filter can be either a finite impulse response (FIR) filter or an infinite impulse response (IIR) filter. The matching filter could be a fixed model or an adaptive model. If the adaptive matching filter is used, the adaptability can be combined with the adaptive filter 30a shown in Fig. 3. The use of the matching filters can be implemented by a digital matching filter in the processor 60 shown in Fig. 5.

In one embodiment, when speech pause detection is used, the weight update of the adaptive filter can be made during pauses in the speech to further reduce the target signal cancellation problem. During pauses in the speech, enhanced noise output contains even fewer speech components.

In one embodiment, the adaptive filter 30a shown in Fig. 3 is a nonlinear filter and can be implemented by a neural network such as a multilayer perceptron network, radial basis function network, high-order neural network, etc. The corresponding learning algorithms in a neural network, such as the back propagation algorithm, can be used for the adaptive filter.

It will be appreciated by those of ordinary skill in the art that the invention can be implemented in other specific forms without departing from the spirit or character thereof. The presently disclosed embodiments are therefore considered in all respects to be illustrative and not restrictive. The scope of the invention is illustrated by the appended claims rather than the foregoing description, and all changes that come within the meaning and range of equivalents thereof are intended to be embraced herein.

Claims:

1. An apparatus comprising:

a fixed rear-null polar pattern unit;

a fixed front-null polar pattern unit; and

a combining unit receiving data from the fixed rear-null polar pattern unit and the fixed front-null polar pattern unit, the combining unit producing reduced noise output, the combining unit including an adaptive filter.

2. The apparatus of Claim 1 wherein the fixed rear-null polar pattern unit, the fixed front-null polar pattern unit, and the combining unit are implemented as a software program running on a processor.

3. The apparatus of Claim 2 wherein the processor is a digital signal processor.

4. The apparatus of Claim 1 wherein the fixed front-null and fixed rear-null polar pattern units produce cardioid polar pattern outputs.

5. The apparatus of Claim 1 wherein the adaptive filter is a finite impulse response filter.

6. The apparatus of Claim 5 wherein the adaptive filter has a single coefficient.

7. The apparatus of Claim 6 wherein the coefficient is limited to the range 0 to -1 so that the null will range from 90° to 180° and 180° to 270°.

8. The apparatus of Claim 1 wherein the output of the apparatus is used to calculate the coefficients for the adaptive filter.

9. The apparatus of Claim 1 wherein the fixed rear-null polar pattern unit and fixed front-null polar pattern unit are implemented using delays and summers.

10. The apparatus of Claim 1 wherein the apparatus comprises a hearing aid.

11. The apparatus of Claim 10 wherein the hearing aid is positionable on the side of a user's head.

12. A method comprising:

calculating signal- enhanced intermediate values from a rear-null polar pattern combination of front and back microphone samples;

calculating noise- enhanced intermediate values from a front-null polar pattern combination of front and back microphone samples;

adaptive filtering the noise- enhanced intermediate values; and

combining the signal- enhanced intermediate values and the filtered noise-enhanced intermediate values to produce a reduced noise output.

13. The method of Claim 12 wherein the method steps are implemented using a digital signal processor.

14. The method of Claim 12 wherein adaptive filtering is done by implementing a finite impulse response filter.

15. The method of Claim 14 wherein the adaptive filter has a single coefficient.

16. The method of Claim 15 wherein the coefficient is limited to the range 0 to -1 so that the null will range from 90° to 180° and 180° to 270°.

17. The method of Claim 12 wherein the adaptive filtering uses filtering coefficients which are modified using the output of the system.

18. The method of Claim 12 wherein the combining step comprises subtracting the filtered noise-enhanced intermediate values from the signal-enhanced values.

19. The method of Claim 12 wherein the rear-null polar pattern and front-null polar pattern are cardioid polar patterns.

20. The method of Claim 12 further comprising the steps of using the reduced noise output to produce a hearing-aid output.

21. A computer-readable medium containing a program which executes the following procedure:

calculating signal- enhanced intermediate values from a rear-null polar pattern combination of front and back microphone samples;

calculating noise- enhanced intermediate values from a front-null polar pattern combination of front and back microphone samples;

adaptive filtering the noise- enhanced intermediate values; and

combining the signal- enhanced intermediate values and the filtered noise-enhanced intermediate values to produce a reduced noise output.

22. The computer-readable medium of Claim 21 wherein the computer-readable memory comprises a memory for a processor.

23. The computer-readable medium of Claim 22 wherein the processor is part of a hearing-aid system.

24. The computer-readable medium of Claim 21 wherein the adaptive filtering includes using the reduced noise output to calculate the adaptive filter coefficients.

25. The computer-readable medium of Claim 21 wherein adaptive filtering is done by implementing a finite impulse response filter.

26. The computer-readable medium of Claim 25 wherein the adaptive filter has a single coefficient.

27. The computer-readable medium of Claim 26 wherein the coefficient is limited to the range 0 to -1 so that the null will range from 90° to 180° and 180° to 270°.

28. The computer-readable medium of Claim 21 wherein the combining step comprises subtracting the filtered noise-enhanced intermediate values from the signal-enhanced intermediate values.

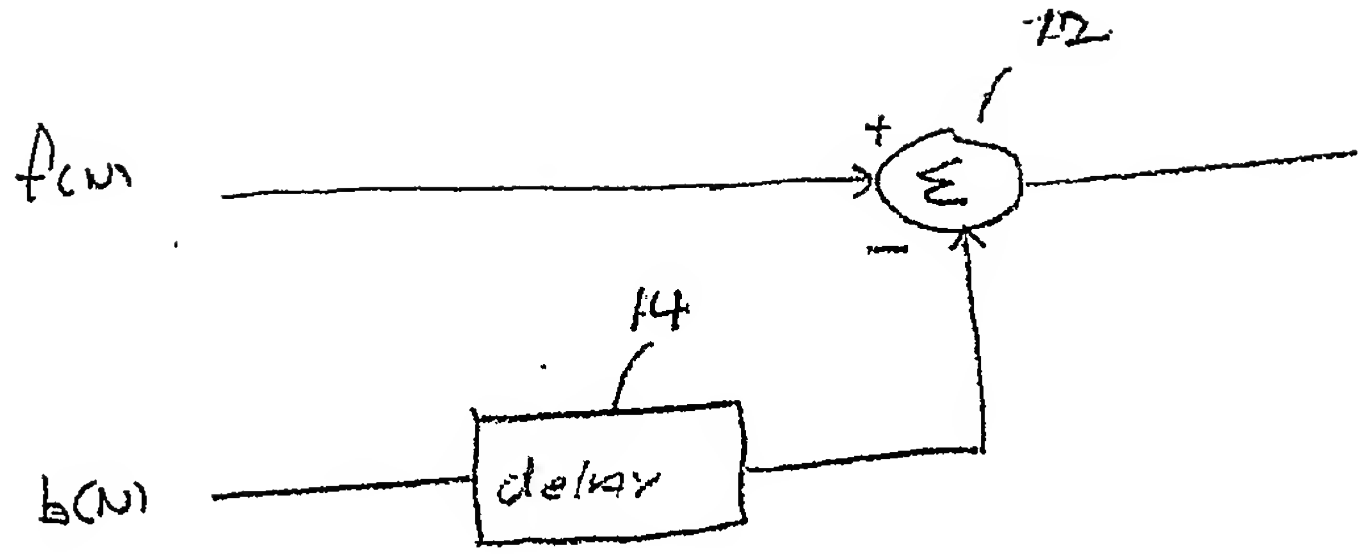

FIGURE 1
(PRIOR ART)

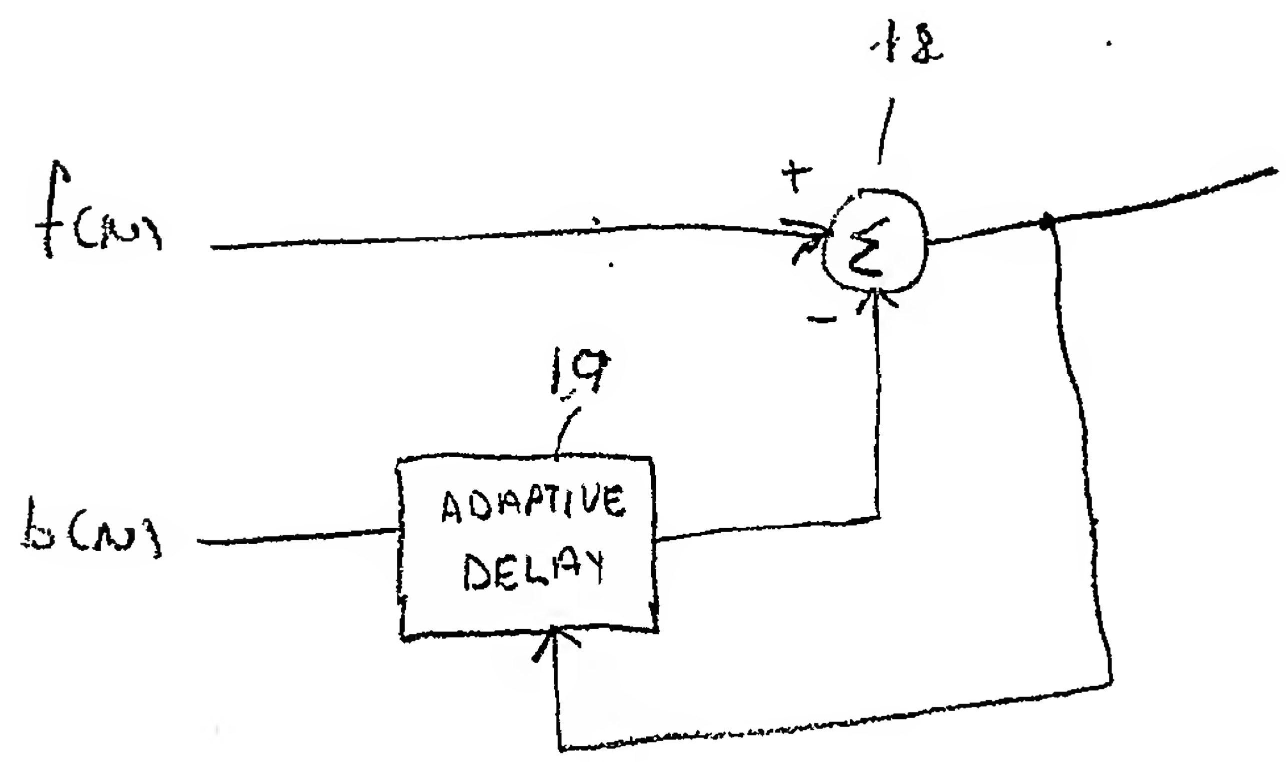

FIGURE 2
(PRIOR ART)

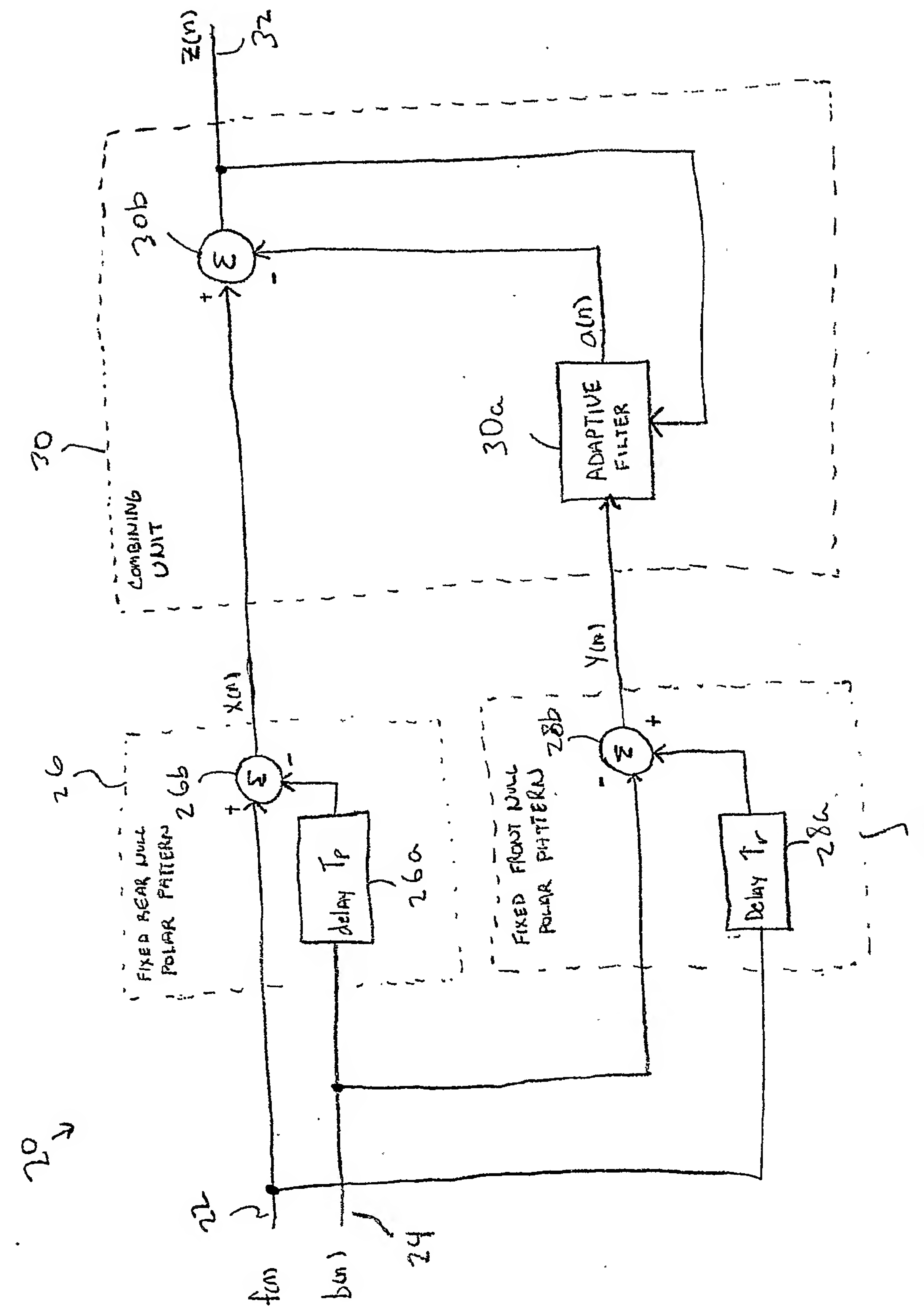
z(n)
32
30b
30
COMBINING UNIT
30a
ADAPTIVE FILTER
a(n)
x(n)
y(n)
26
FIXED REAR NULL POLAR PATTERN
26b
delay Tp
26a
FIXED FRONT NULL POLAR PATTERN
28b
Delay Tr
28a
28
20
22
24
f(n)
b(n)

FIGURE 3

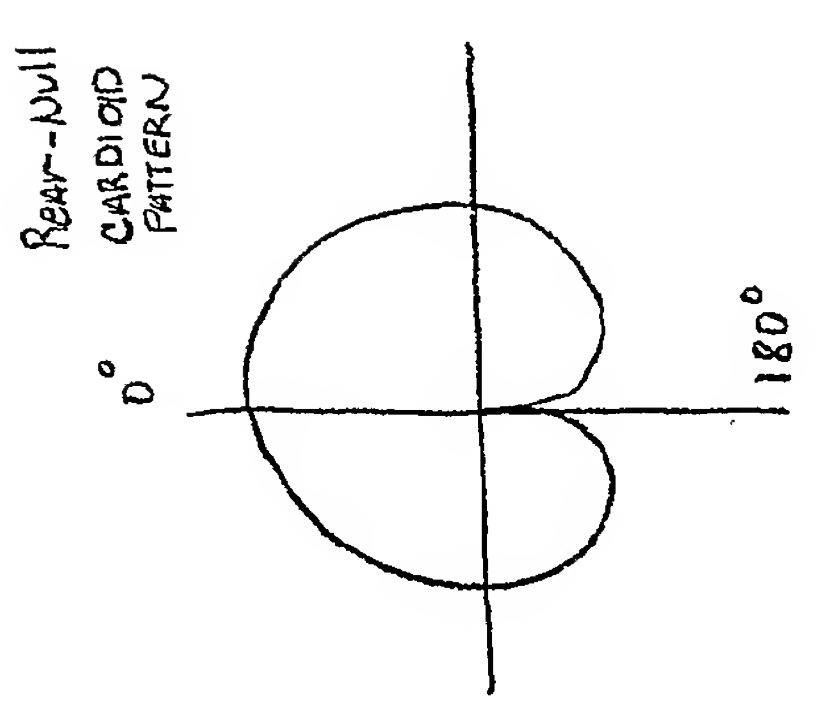

FIGURE 4A

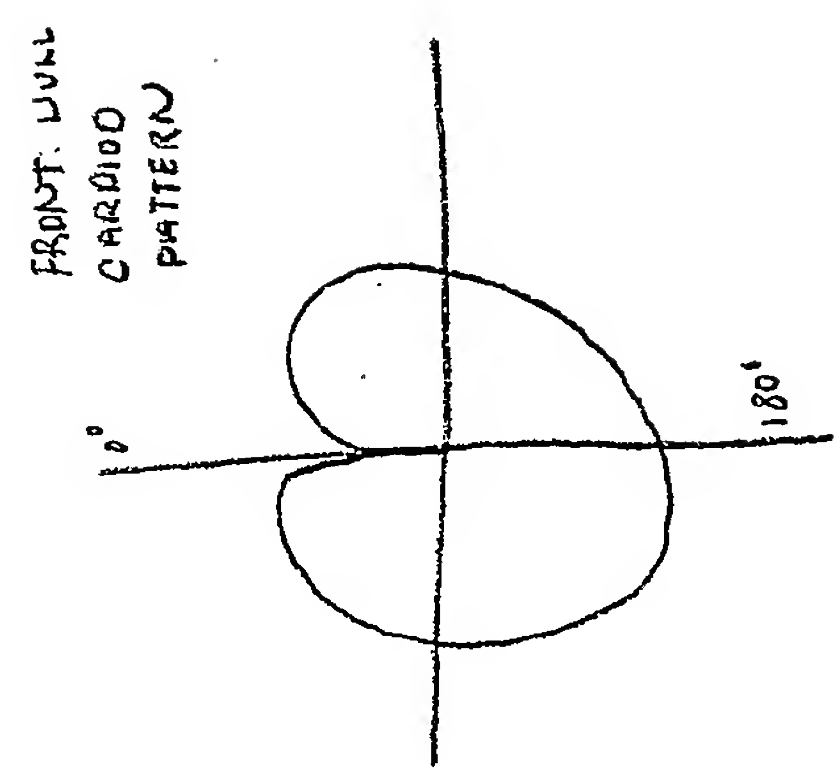

FIGURE 4B

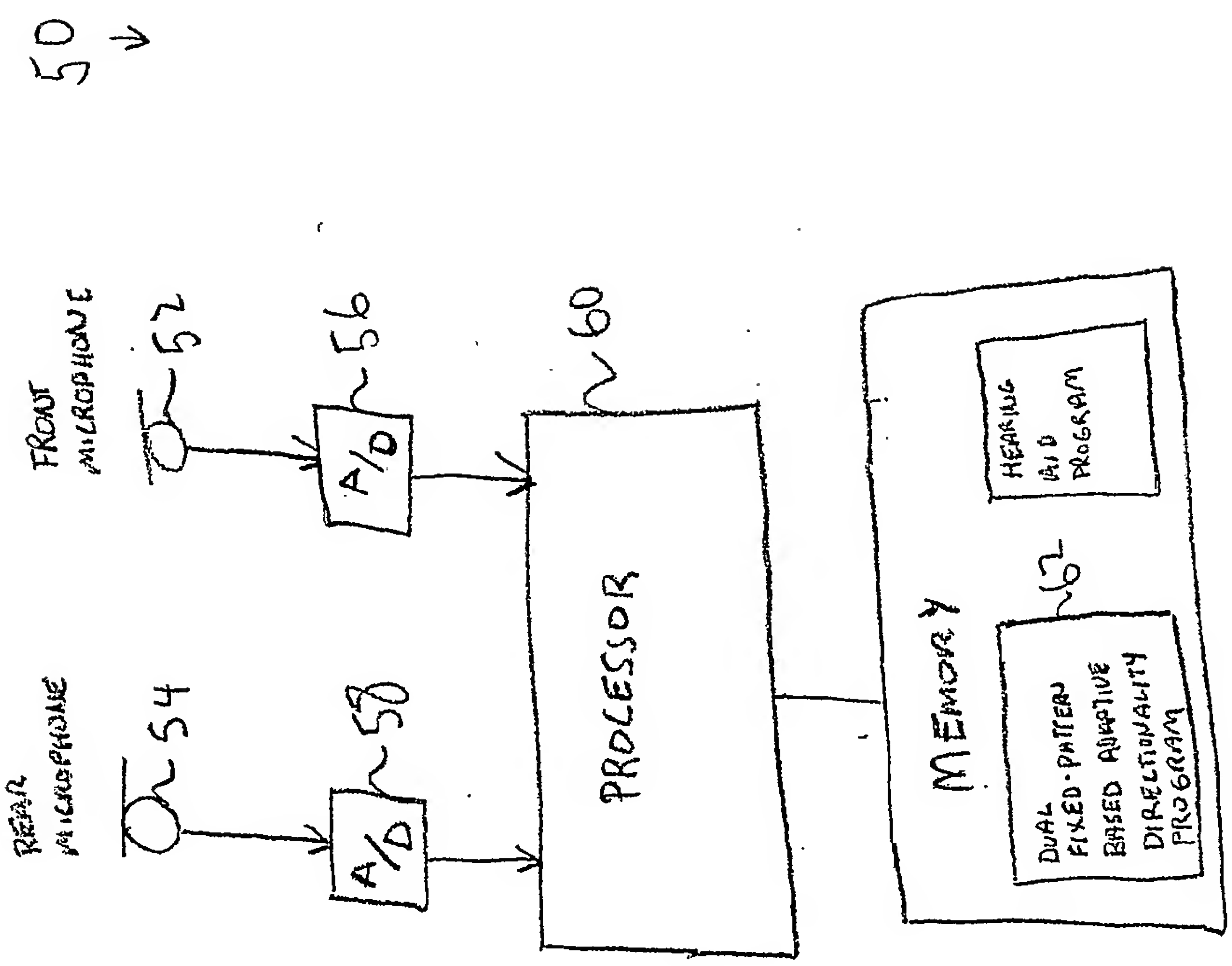
50
FRONT MICROPHONE
52
REAR MICROPHONE
54
A/D
56
A/D
58
PROCESSOR
60
MEMORY
HEARING AID PROGRAM
62
DUAL FIXED-PATTERN BASED ADAPTIVE DIRECTIONALITY PROGRAM

FIGURE 5

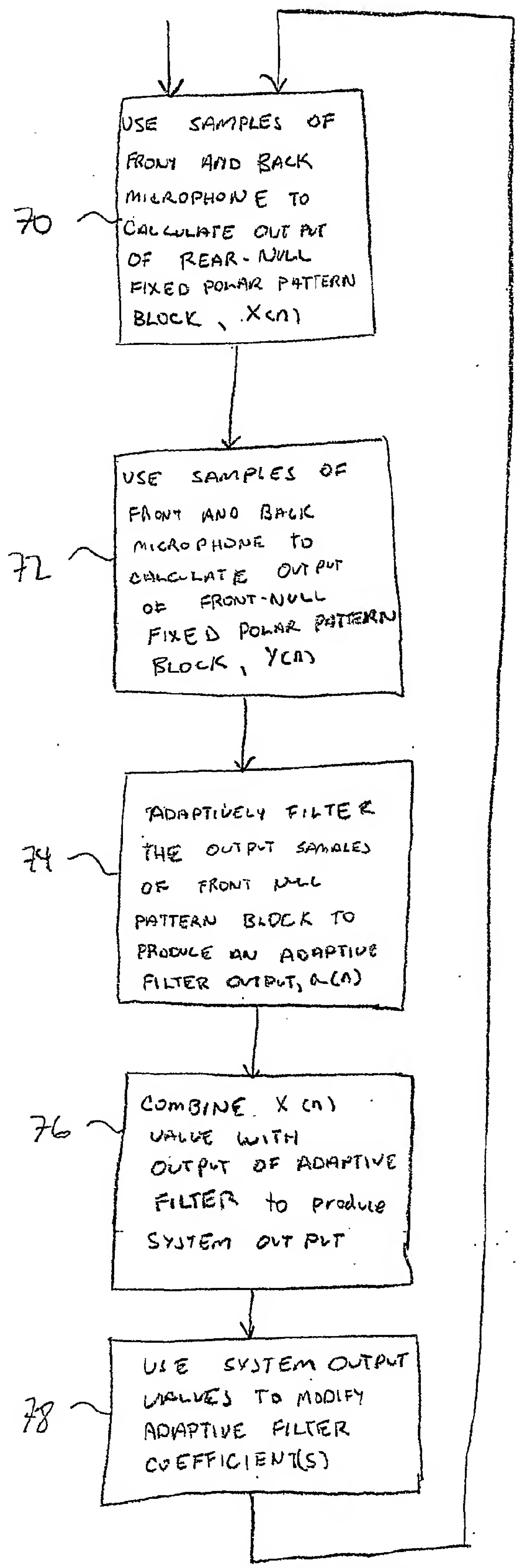
USE SAMPLES OF FRONT AND BACK MICROPHONE TO CALCULATE OUTPUT OF REAR-NULL FIXED POLAR PATTERN BLOCK, X(n)
70
USE SAMPLES OF FRONT AND BACK MICROPHONE TO CALCULATE OUTPUT OF FRONT-NULL FIXED POLAR PATTERN BLOCK, Y(n)
72
ADAPTIVELY FILTER THE OUTPUT SAMPLES OF FRONT NULL PATTERN BLOCK TO PRODUCE AN ADAPTIVE FILTER OUTPUT, a(n)
74
COMBINE X(n) VALUE WITH OUTPUT OF ADAPTIVE FILTER to produce SYSTEM OUTPUT
76
USE SYSTEM OUTPUT VALUES TO MODIFY ADAPTIVE FILTER COEFFICIENT(S)
78

FIGURE 6